IN THE SPECIFICATION:

Please replace the paragraph beginning at page 25, line 12, with the following rewritten paragraph:

--The internal operation of scalable call processing node **200** illustrated in Figure 6 will now be explained with reference to the flow chart illustrated in Figure [[9]] 10. In Figure [[9]] 10, in step **ST1**, LIM **201** receives an ISUP message. Such a message may be an initial address message (IAM), an address complete message (ACM), an answer message (ANM), a release message (REL), or a release complete message (RLC). In this example, it is assumed that an IAM message is received. In step **ST2**, LIM **201** illustrated in Figure 6 determines whether the message should be through-switched. As stated above, this determination may be made based on the destination point code in the message. In step **ST3A**, if the message is to be through-switched, HMDC process **602** in LIM **201** routes the message to the appropriate module for outbound processing. In this example, it is assumed that the message is not a message that is to be through-switched.--

Please replace the paragraph beginning at page 26, line 8, with the following rewritten paragraph:

--An additional function that may be performed by call processor **604** is determining whether translation is required. As used herein, translation refers to translation to or from a normalized ISUP protocol. In order to make this determination, call processor **604** may determine the ISUP ~~message~~ protocol used by the called party end office based on one or more parameters, such as DPC_1 in the received ISUP message. In step **ST8**, if translation is required, call processor **604** may forward the

message to ISUP translator **607**, where a translation is performed, and receive a translated message from translator **607**.--

Please replace the paragraph beginning at page 30, line 6, with the following rewritten paragraph:

--In response to the SIP message, MGC **210** generates the CreateConnection message requesting MG **806** to set up a trunk connecting point code 2-1-1 and point code 55-2-2. Thus, the embodiment in Figure 12 illustrates call setup using SIP according to an embodiment of the present invention.--